International Journal of Engineering Sciences & Emerging Technologies, August 2012.
ISSN: 2231 – 6604 Volume 3, Issue 1, pp: 9-21 ©IJESET

COMPARATIVE PERFORMANCE OF HIGH SPEED NETWORKS CARRYING MULTIMEDIA

Ehab Aziz Khalil
Department of Computer Science & Engineering, Faculty of Electronics Engineering,
Menoufiya University, Menouf-32952, EGYPT
drehab_khalil@yahoo.com & drehab.khalil@gamil.com

ABSTRACT

All the early access mechanism based on contention or sequential access does not scale well to multi-gigabit per second rates or larger geographical coverage. Advances in laser and fiber optic technology have driven feasible link throughputs above a gigabit per second. Dynamic RAM chip have become cheap enough to be cost effective at providing the large amounts of buffering needed at these very high link speeds. Moreover, quick routing and MAC decisions are possible with current CMOS technology. In combination, these trends make it possible to construct a Practical LAN using multiple switches and gigabit per second point-to-point fiber links configured in an arbitrary topology. It's well known that the current technology trends make it possible to build high speed communication networks that can support high performance distributed computing. With increased data rates, the packet transmission time of a network could approach or even become less than the medium propagation delay. Performance of many network schemes degrades rapidly when the packet transmission time become small compared to the medium propagation delay. The use of high speed data rate alone is not sufficient, unless the medium access provides for efficient bandwidth use of the medium. This paper discusses and describes the performance characteristics of the most popular high speed networks such as DQDB, FDDI ,and ATM for Broadband Integrated Service Digital Network (B-ISDN), while carrying multimedia traffics and compare the performance evaluation results of the these networks with the results of the proposed high speed network namely EHABKIT. The comparative of the results confirms that the concept of pure packet switching for the EHABKIT architecture is almost equivalent to the high speed networks, DQDB, FDDI, ATM for B-ISDN, and its promising enough. .

KEYWORDS: *Comparative performance of high speed networks carrying multimedia:, DQDB, FDDI, ATM for Broadband ISDN (B-ISDN), Cell Relay, SONET / SDH, DATAKIT packet switch, and EHABKIT architecture.*

I. INTRODUCTION

The term high speed network covers many means of communication. The progress achieved in networks has had a considerable effect on communication between terminals and computers. Over limited distances these networks provide high speed support which meets the requirements of transfer speed and access time. With large-scale organizations the interconnection of establishments usually uses the public domain, where speeds remain more modest. However, recent developments in long distance communication hardware allow hope that speeds will ultimately be harmonized, permitting a performance independent of the location of the terminal.

In most cases the technical principles remains the same, and it is important that the existing ones are well understood in order to appreciate these new networks [1]. Advances in laser and fiber optic technology have driven feasible link throughputs above a gigabit per second. Dynamic RAM chips have become cheap enough to be cost-effective at providing the large amount of buffering needed at these very high link speeds. Moreover, quick routing and MAC decisions are possible with current CMOS technology [2].

In combination, these trends make it possible to construct a practical LAN using multiple switches and gigabit per second point-to-point fiber links configured in an arbitrary topology. This kind of a network has several advantages [3]. In contrast to networks like ETHERNET [4], that use a broadcast physical medium, or networks like FDDI [5,6], based on a token ring, arbitrary topology point-to-point network offer:

1-Aggregate network bandwidth that can be much larger than the throughput of a single link,

2-The ability to add throughput incrementally by adding extra switches and links to match worked requirements,

3-The potential for achieving lower latency, both by shortening path lengths and by eliminating the need to acquire control over the entire network before transmitting,

4-A more flexible approach to high availability using multiple redundant paths between hosts.

High performance networks have the potential to change the nature of distributed computing. Low latency and high throughput communication allow a much closer coupling of distributed systems than has been feasible in the past with previous generation networks, the high cost of sending messages led programmers to carefully minimize the amount of network communication [7]. Further when combined with today's transfer processors, faster networks can enable a new set of applications, such as desktop multimedia and the use of a network of workstations as a supercomputers.

A primary barrier to building high performance networks is the difficulty of high speed network of taking data arriving on an input link of a network and quickly sending it out on the appropriate input link. The networking task is simplified if the data can processed in fixed length packets. Given fixed length packets, networking involves at least two separate tasks [2]:

1- *Scheduling*: choosing which packet to send during each time slot, when more than one packet is destined for the same output, and
2- *Data forwarding*: delivering the packet to the output once it has been scheduled.

In general, we can say that there are two reasons need for high speed networks [8]:

(a) a dramatic increase in computer processing power over the last few years, and
(b) an enormous increase in the volume of stored and processed data.

As a result, a lower speed network may well become the bottleneck between devices needing to transfer large amounts of data with minimal delay.

The remaining of the paper organized as the following, section 2 presents some of the related works. Section 3 gives brief overview of high speed networks. The simulation results explores in section 4. Section 5 produces a comparison and discussion between the explained networks. Finally section 6 presents the conclusion.

II. RELATED WORKS

It is to be noted that most of the bandwidth in current and future high speed networks would be taken up by the multimedia applications, transmitting digital audio and video. Traditional networking protocols are not suitable for this as they do not provide guaranteed bandwidth, end-to-end delay or delay jitter, nor do they have addressing schemes or routing algorithms for multicast connections. High speed networking for multimedia applications is a collection of high quality research papers which address these issues, providing interesting and innovative solutions. It is an essential reference for engineers and computer scientists working in this area. It is also a comprehensive text for graduate students of high-speed networking and multimedia applications. Recently, a performance comparison of video multicast of the two technologies is described in [9]. In [10], a new dynamic bandwidth allocation scheme called Minimum Overflow Traffic Algorithm (MOTA) is proposed to assign the bandwidth for each traffic class in the hierarchical admission control structure in ATM network. In [11], study the problem of bandwidth allocation for ATM network loaded with real-time VBR compressed video traffic. The authors in [12] describes on the research of the process flow and analysis of voice over asynchronous transfer mode in a common communication network. Specifically this project focuses on the process flow of the voice transmission via am ATM network and a simple experiment to build a VoATM network in a laboratory. Study some of the ATM-compatible multiple access MAN (Metropolitan Area Network) protocols, including DQDB (Distributed Queue Dual Bus, also known as IEEE 802.6), CRMA (Cyclic Reservation Multiple Access), DQMA (Distributed

Queue Multiple Access), and FDQ (Fair Distributed Queue), and their performance for different kinds of multimedia application, including interactive education, on-line medical information system, and video conferences discussed in [13]. In [14], a case of a bridged FDDI network carrying multimedia traffic is studied in detail. Also, the details of how to adjust the TTRT and how the adjustment affects the behaviors of FDDI ring are discussed and presented in [15]. The performance of FDDI networks in terms of their guarantee probability, i.e., the probability that a set of synchronous messages are guaranteed to meet their deadlines is explained in [16]. A mathematical model for a fiber distributed data interface (FDDI) network with multimedia voice and data traffic are developed, and the performance of the network is obtained by approximate queuing analysis and simulations presented in [17]. The performance of two high speed network protocols, IEEE 802.6 distributed queue dual bus MAN and fiber distributed data interface are compared based on their ability to support integrated traffic has been discussed in [18]. A comparison of the MAC access protocols of the IEEE 802.6 DQDB MAN and FDDI standards is presented in [19]. A DATAKIT packet switch demonstrates how the byte-stream concept can integrate local area and wide area network objectives, more details in [20].

III. Brief Overview of High Speed Networks

In this section, shortly details of the protocols relevant to our study. Complete descriptions of these protocols, including the handling of various error conditions may be found in the papers cited.

3.1 Distributed Queuing

The purpose of distributed queuing is to obtain, or at least to approximate, a single view of a first-come-first-service (FCFS) queue for each transmission in all active node across the network. The underlying network is a dual bus consisting of unidirectional slotted buses A and B operating in opposite directions. The slots are generated by the header node of each bus. Every node receives and transmits on both buses, so bus selection is based on the destination. Reservations for transmission on bus A are made in bus B via requests and vice versa. Since the operation of both buses is identical, we'll consider transmission on bus A.

The **Distributed Queue Dual Bus (DQDB)** [21,22], protocol reserves a slot on bus A via a request bit in a slot on bus B as shown in Figure 1.

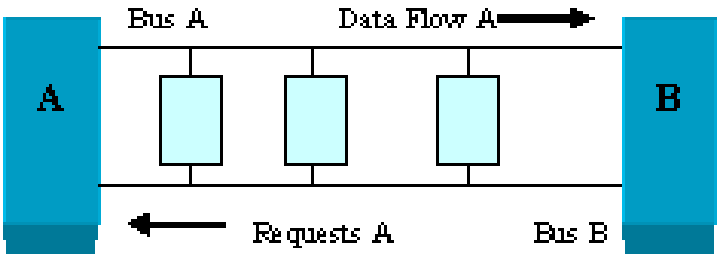

Figure 1. Distributed Queue Dual Bus Architecture

Access to the bus is controlled by a request and countdown counter. The request counter keeps track of the current free-slot requirements of downstream nodes whereas the content of the countdown counter indicates the number of free-slots to be passed before the node's own transmission takes place. Transmissions are scheduled one at a time. Scheduling is done by transferring the contents of the request counter to the countdown counter, resetting the request counter, and initiating the transmission of a request on the reverse bus. In the ideal case, the newly scheduled transmission is therefore queued behind all known pending slot transmission of the downstream nodes. It should be noted that a node can schedule its next transmission as soon as the previous one has been transmitted. Thus, there is no need to wait for the reservations themselves. Furthermore, an outstanding request counter can cumulate requests during periods when request bits in passing slots on the opposite bus are already set, so that data transmission can be made before requests are transmitted. This protocol has evolved into the *IEEE802.6* standard [22]. Good bibliography reference on DQDB is in [23].

3.2 Multiple Packets and Destination Removal

The **Fiber Distributed Data Transfer (FDDI)** [24], has become a high speed Network standard [25], and provides a data rate of 100-Mbps. The basic topology is a dual (counter–rotating) ring consisting of a primary ring for data transfer and a secondary ring for redundancy. Access to the ring is by a timed-token mechanism similar to that used for the Token Bus standard. Distinctions with respect to the access mechanism are that FDDI foresees eight instead of four priorities and that a single token-rotation time is measured between two node visits, whereas in the Token Bus scheme the measurement is done for each of the three lower priority levels.

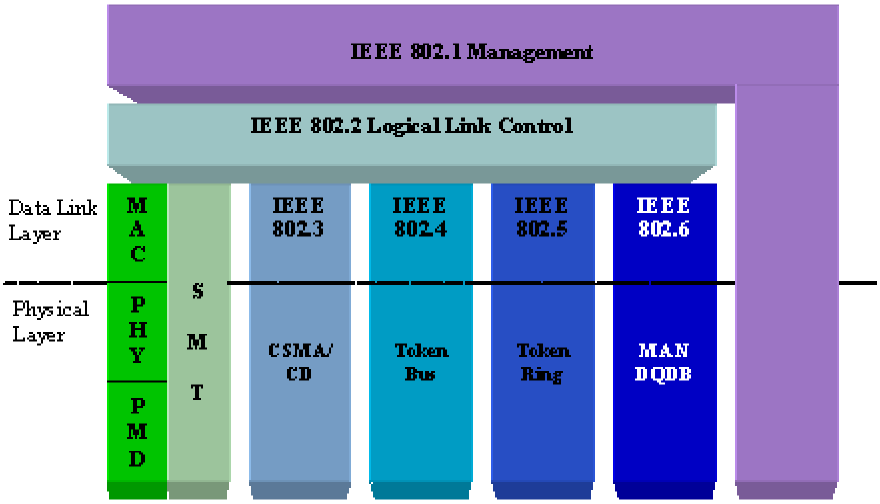

Figure 2. DDI Protocol in the ISO Reference Model

As shown in Figure 2, the FDDI documents cover the Physical Layer and the MAC sub-layer LANs to higher protocols. Thus, the transition from lower speed LANs to FDDI as well as the interconnection of LANs by a FDDI backbone should be achieved [24,26]. In FDDI-II, part of the transmission capacity can be assigned to isochronous channels which support strict timing services like circuit-switched telephony and video. The remaining part for data and is controlled by the timed-token protocol.

3.3 Broadband ISDN, Cell Relay, and SONET

1- **B-ISDN Architecture** [27], the Broadband Integrated Service Digital Network (B-ISDN) differ from a narrowband ISDN in a number of ways. To meet the requirement for high-resolution video, an upper-channel rate on the order of 150Mbps is needed. To simultaneously support one more interactive services and distributive services, a total subscriber line rate of about 600-Mbps is needed. In terms of today's installed telephone plants, this is a stupendous data rate to sustain. The only appropriate technology for widespread support of such data rates is optical fiber. Hence, the introduction of B-ISDN depends on the pace of introduction of fiber subscriber loops. Internal to the network, there is the issue of the switching technique to be used. The switching facility has to be capable of handling a wide range of different bit rates and traffic parameters (e.g., burstiness). Despite the increasing power of digital circuit-switching hardware and the increasing use of optical-fiber trucking, it is difficult to handle the large and divers requirements of B-ISDN with circuit-switching technology. For this reason, there is increasing interest in some type of fast packet switching as the basic switching technique for B-ISDN. This form of switching readily supports a new user-network interface protocol known as Asynchronous Transfer Mode (ATM).

Broadband ISDN Protocols: The protocol architecture for B-ISDN introduces some new elements found in the ISDN architecture, as depicted in Figure 3.

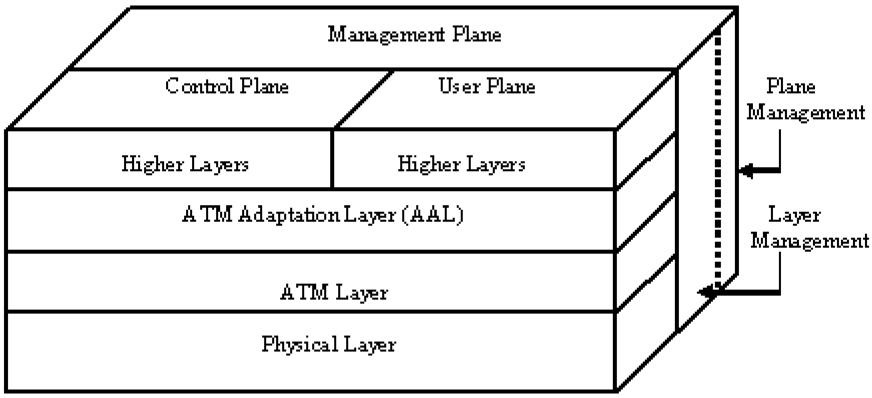

Figure 3. B-ISDN Protocol Reference Model

For B-ISDN, it is assumed that the transfer of information across the user-network interface will employ what is referred to as Asynchronous Transfer Mode (ATM) [28,29]. ATM is essence, a form of packet transmission across the user-network interface in the same way that X.25 is a form of packet transmission across the user-interface. One difference between X.25 and ATM is that includes control signaling on the same channel as data transfer, whereas ATM makes use of common-channel signaling. Another difference is that X.25 packet may be varying length, whereas ATM packets are fixed size, referred to as cells.

The decision to use ATM for B-ISDN is a remarkable one. This implies that B-ISDN will be a packet based network, certainly at the interface and almost certainly in terms of its internal switching. Although the recommendation also states that B-ISDN will support circuit-mode applications, this will be done over a packet-based transport mechanism. Thus, ISDN, which began as an evolution from the circuit-switching telephone network, will transform itself into a packet-switching network as it takes on broadband services. The protocol reference model makes reference to three separate planes:

(1) *User Plane*: provides for user information transfer, along with associated controls (e.g., flow control, error control).
(2) *Control Plane*: performs call-control and connection-control functions.
(3) *Management Plane*: includes plane management, which perform management functions related to a system as a whole and provides coordination between all planes, and layer management, which performs management functions related to resources and parameters residing in its protocol entities.

2-Cell Relay [27], Cell relay, also known as Asynchronous Transfer Mode, is, in a sense, a culmination of all the developments in circuit switching and packet switching over the past 20 years. One useful way to view cell relay is as an evolution from frame relay (packet switching). Both frame relay and ATM takes advantage of the reliability and fidelity of modern digital facilities to provide faster packet switching than X.25. Like frame relay and X.25, cell relay allows multiple logical connections to be multiplexed over a single physical interface. As with frame relay, there is no-link error control or flow control with cell-relay. The most obvious difference between cell-relay and frame-relay variable-length packets, called frames, and cell-relay uses fixed length packets, called cells. As with frame-relay, cell-relay provides minimum overhead of error control, depending on the inherent reliability of the transmission system and on higher layers of logic to catch and correct remaining errors. By using a fixed packet length, the processing overhead is reduced even further for cell-relay compared to frame-relay. The result is designed to work in the range of tens and hundreds of megabits per second, compared to the 2Mbps of frame relay. Another way to view cell-relay is as an evolution from multi-rate circuit switching – with multi-rate circuit switching only fixed data rate channels are available to the end system. Cell-relay allows the definition of virtual channels with data rates that are dynamically defined at the time that virtual channel is created. By using small, fixed-size cells, cell-relay is so efficient that it can offer a constant data rate channel even though it is using a packet switching technique. Thus, cell-relay extends multi-rate circuit switching to allow multiple channels with data rate of each channel dynamically set on demand. With cell-relay, a key design issue is the size of the cell. The choice of cell size is governed by a trade–off among factors such as overall network delay, transmission efficiency, and network complexity. Depending on the weighting given to the various factors, a payload value for cell-relay is between 32 and 64 octets is derived [30].

3-SONET/SDH [27], SONET (Synchronous Optical NETwork) is an optical transmission interface originally proposed by Bell Core and standardized by ANSI. A compatible version, referred to as Synchronous Digital Hierarchy (SDH), has been published by CCITT in Recommendation G.707, G.708, and G.709 [17]. SONET is intended to provide a specification for taking capability of optical fiber. The SONET standard addresses the following issues [31]: (1) It establishes a standard multiplexing format using any number of 15 - 84 Mbps signals as building blocks. Because each building can carry a D53 signal, a standard rate is defined for any high bandwidth transmission system that might be developed. (2) It establishes an optical-signal standard for interconnecting equipment from different suppliers. (3) It establishes extensive cooperation, administration, and maintenance (DAM) capabilities as part of the standard. (4) It defines a synchronous-multiplexing format for carrying lower- level digital (DS1, DS2, CCITT standards). The synchronous structure greatly simplifies the interface to digital switches, digital cross-connect switches and add-drop multiplexers. (5) It establishes a flexible architecture capable of accommodating future applications such as B-ISDN with a variety of transmission rates.

The key requirements have driven the development of SONET. First was the need to push multiplexing standards beyond the existing DS3 (44.736Mbps) level. With the increasing use of optical transmission systems, a number of vendors have introduced their own proprietary schemes of combining anywhere from two twelve DS3s into an optical signal. In addition, the European schemes, based on the CCITT hierarchy are incompatible with North American schemes, SONET provides a standardized hierarchy of multiplexed digital-transmission rates that accommodates existing North American and CCITT rates.

A second requirement was to provide economic access to small amount of traffic within the bulk of an optical signal. For this purpose, SONET introduces a new approach to time-division multiplexing.

A third requirement is to prepare for future sophisticated services offerings, such as virtual private networking, time-of-delay bandwidth allocation, and support of the B-ISDN ATM transmission technique. To meet this requirement, a major increase in network management capabilities within the synchronous time division signal was needed.

4- EHABKIT a Proposed High Speed Network [32-36], **EHABKIT** is a high-speed, and high performance dual bus multiple access network. It is a single integrated network to used as a LAN/MAN, and WAN [33]. Furthermore it can also make use of a twisted pair, optical fiber building distribution system that both complements and enhances existing building wiring. The motivation for the design is from **DATAKIT** (**DATAKIT** was designed at **Bell Labs** and is currently a registered trademark of **AT&T Labs**.) packet switch due to Fraser [37].

The Modified version for what is referred to as **EHABKIT** (**E**nhanced **H**igh speed **A**rchitecture **B**us **KIT**) an extension to the earlier work [38,39]. **EHABKIT** based upon a currently located short high-speed buses [32], therefore **EHABKIT** enjoys the fundamental advantages of bus topology and short high-speed buses. With the bus topology, the communication network is simply, the transmission medium no switches, and no repeaters. All stations attach, through appropriate hardware interfacing directly to a linear transmission medium or bus. The short high-speed enjoys a fundamental advantage relative to distributed approaches in that the propagation delay along the length of the bus is smaller than the transmission time of a single bit, even at data rates of tens of megabits per second. This feature can be exploited to provide simplicity efficiency in accessing the transmission medium. **EHABKIT** permits multiple priority traffic of classes with fair allocation of bandwidth within each class [32-36], along with a capability of integrated circuit and packet switching. **EHABKIT** demonstrates how byte-stream concept can integrate LAN and WAN objectives. It is expected to support a wide range of applications through short high-speed and flexible multimedia communication capabilities. Multimedia includes voice, data, moving picture (video), images, graphics, [32-36].

This is evident from the capacity can support **225 to 450 voice lines** [38], which compares will with the **IEEE802.5 Token Ring**, and it is also close to the capacity of **100Mbps FDDI Token Ring** which can support maximum of **500 station** on a network. **EHABKIT** is ensure fair sharing of the transmission capacity, obtain high throughput and network utilization together with low and bound delays, support priorities and different traffic classes, strive for simplicity, robustness easy implement ability, all at the same time. In fact these are the challenge of the proposed Media Access Control (MAC) protocol with a centralized scheduling. **EHABKIT** is an assembly of Interface Cards (**ICs**), connected by a dual of short high-speed buses. Each type of IC serves one type of remote equipment and, if need be, terminates the protocol of that equipment. There are ICs for trunks **NIC** (Network Interface Card), that lead to other **EHABKITs** for terminals, for host computers. There modules provide **EHABKIT** timing, network control and maintenance support named by Clock Module (**CM**), and Supervisor Module (**SM**), as shown in Figure 4.

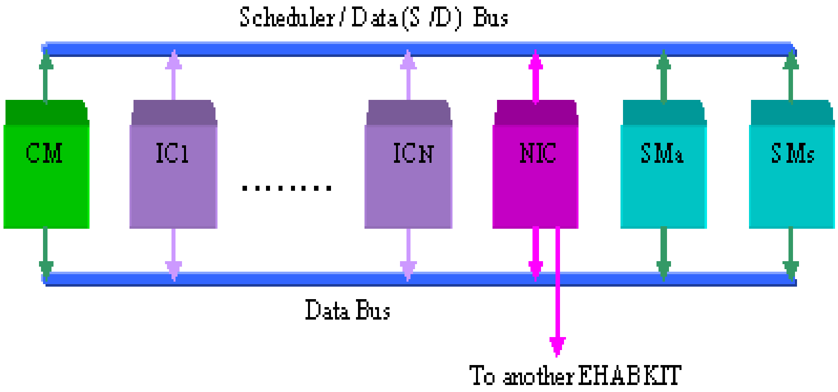

Figure 4. Block Diagram of EHABKIT Architecture

IV. Simulation Results

In our study [40,41] we've proposed an efficient FDDI's MAC (Media Access Control) for increasing both the throughput of asynchronous traffic and the capacity of synchronous sources. The results (see Table: 1, Figure 5 [42] and Figure 6 [43,44]) confirm that the proposed MAC provides FDDI more efficient for supporting multimedia applications.

Table: 1.

Case Study	Standard FDDI's MAC	Modified FDDI's MAC	
		α = 0.5	**α = 0.25**
Maximum Number of Video Source (*Nv*)	**34**	**33**	**50**
Asynchronous Traffic Throughput	**44.75**	**48.4**	**22.8**

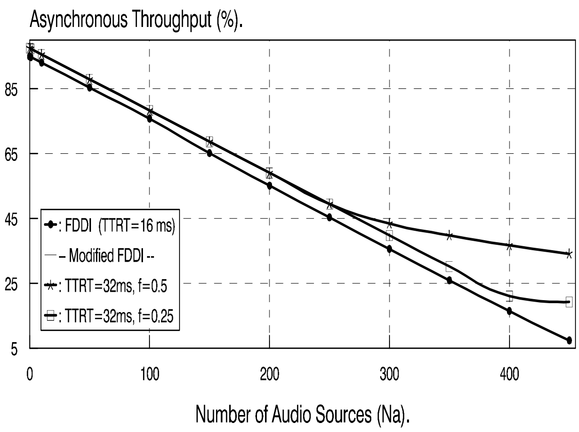

Figure 5. Throughput Characteristics of Asynchronous Traffic Versus *Na* (FDDI & Modified FDDI)

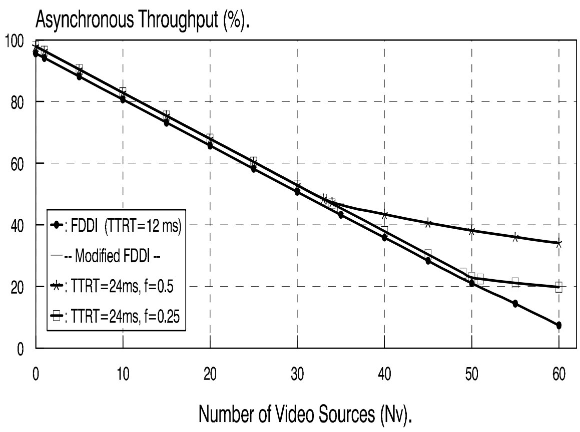

Figure 6. Throughput Characteristics of Asynchronous Traffic Versus *Nv* (FDDI & Modified FDDI)

The following Table: 2 [45,46] shows the maximum number of video sources (*Nv*) and the average delay of video packets (*Davg*) for the case when the video rate is equal to 1.5Mbps and the allowable delay is equal to *30 msec*. From the table it is clear that the video packet length (*Pv*) between 1125-2625 bytes achieve the maximum number of video sources *Nv* = 63 source [43]. Obviously, the shorter video packet length, the smaller average network delay.

Table: 2.

Video Packet Length (Byte)	Maximum Number of Video Sources (*Nv*)	Average Video Delay (*Davg*) *ms*
375	**60**	**10.11**
750	**62**	**11.72**
1125	**63**	**14.72**
1500	**63**	**15.73**
2250	**63**	**18.46**
2625	**63**	**21.31**
3000	**54**	**23.27**
3375	**40**	**24.18**

Table: 3 summaries the simulation values at two cases: integration of video / voice and the integration of data/video/voice over ATM [28,29,44, 47-50]. In the table note that MWT= Mean Waiting Time. Figure 10 illustrates video MBS and voice MBS versus N_{vo}, and Figure 11 illustrates voice MBS and data MBS versus N_{vo}, for R_{vi}= 1.5 Mbps, R_{vo}=192 Kbps, N_{vi} sets to various values such as 10 and 20 sources. M_{siz} sets to 100 and 300 cells, with μ = 5 ms. Figure 7 and 8, indicate that with the increasing of N_{vo} the voice MBS slightly increases, video MBS and data MBS approximately remain constant up to saturation limit, beyond the saturation limit, video, voice, and data MBS sharply increase because of the increasing of N_{vo} increases the number of voice cells, resulting in increases in video, voice, and data MBS. The video MBS remains constant for short interval and then increases up to saturation limit. That is because the ratio of each traffic is suddenly change (N.B. the ratio of each traffic ignore the fraction so the ratio increment or decrement suddenly). Table 3 shows the different fixed generated rates of the video/data integration, and corresponding N_{vo}, video MBS, voice MBS, and data MBS.

Table: 3.

Video Rate=1.5Mbps, Voice Rate=192Kbps, Mean arrival time of data μ= 5ms					
Voice Sources=100, Video Source=10			**Voice Sources=50, Video Source=10**		
Message Size (Cell)	**Video MWT (Cell)**	**Voice MWT (Cell)**	**Message Size (Cell)**	**Video MWT (Cell)**	**Voice MWT (Cell)**
0	**30**	**148**	**0**	**30**	**82**
100	**32**	**156**	**100**	**30**	**90**
300	**34**	**236**	**300**	**39**	**144**

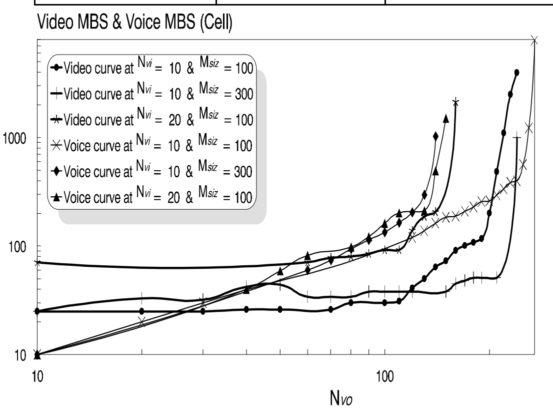

Figure 7. depicts video MBS & voice MBS versus N_{vo} with R_{vi} =1.5 Mbps and R_{vo} = 192 Kbps and μ = 5 ms.

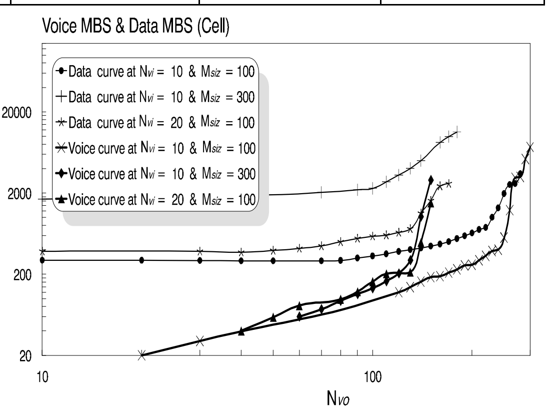

Figure 8. depicts voice MBS & data MBS versus N_{vo} with R_{vi} =1.5 Mbps and R_{vo} = 192 Kbps and μ = 5 ms.

In the proposed architecture the real-time traffic has a higher priority transmission and it is with a good acceptable delay about 10ms at heavy loads in case of Free Round Robin (FRR) [32-36] and less than 250ms in case of Restricted Voice Data Round Robin (RVDRR) [32-36] as shown in the Table: 4, (in the following Table Note that NT mean Non Transmission). From Figure 9, 10, it is clear that the capacity can support 225 to 450 voice lines, which compares well with the DATAKIT which can support 380 asynchronous terminals.

Table: 4

Number of IC=45, Number of Devices per IC=6, λ=170ms, μ=410ms, and Mean Time Between Arrival (MTBA) =60									
Voice Packet Length (Bits)	Average Delays (ms)								
	FRR Service			RVRR Service			RVDRR Service		
	Voice	ID	NID	Voice	ID	NID	Voice	ID	NID
128	32.82	NT	NT	103.14	42.37	NT	75.85	45.38	49.54
256	9.43	193.35	NT	138.26	63.71	NT	118.33	96.49	104.49
384	9.55	188.97	NT	165.54	80.04	NT	147.84	135.71	148.43
512	11.92	212.76	NT	192.14	103.97	NT	177.07	170.06	187.83
640	14.51	224.36	NT	215.68	127.27	NT	205.92	215.23	234.29
768	17.01	250.92	NT	225.41	141.73	NT	229.42	259.70	281.29
896	19.51	255.40	NT	232.98	161.91	NT	249.60	290.62	316.34

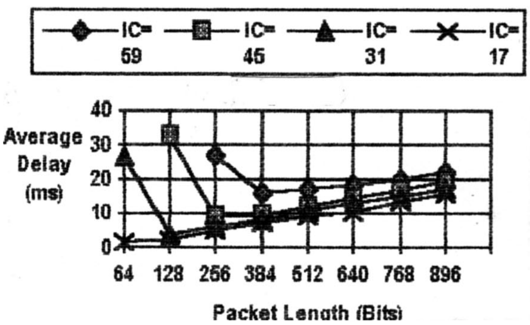

Figure 9. Packet length Verses Average Delay.

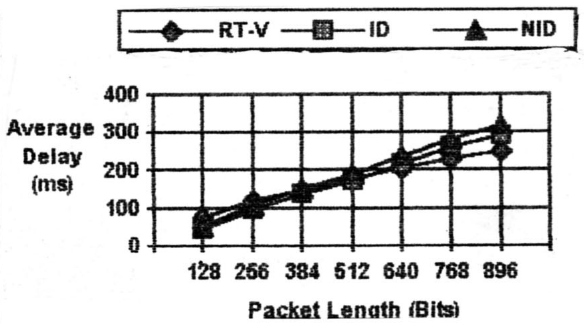

Figure 10. Packet length Verses Average Delay.

V. COMPARISON AND DISCUSSION BETWEEN PREVIOUS NETWORKS

This section presents a comparison of the proposed EHABKIT architecture with some other media access schemes. In Distributed Queue Dual Bus (DQDB) and its variations [51], a head station generates the frame synchronization on one bus, and the end station generates the frame pattern at the same rate on the other bus. In the proposed EHABKIT architecture only one module named by the clock module (CM) maintains and synchronous time-slots frames for both scheduler slot and data cycle on the dual bus (S/D bus and data bus). In DQDB, priority operation can be achieved by having separate distributed queues for each level of priority. The proposed architecture favorably compares with this too. At high a values (i.e., with a large end-to-end propagation delay and a comparatively small slot size) and high load, the DQDB protocol does not appear to fair in its bandwidth allocation. Several papers discuss how the DQDB protocol could behave in an unfair manner (good survey is in [8,21]). In the proposed architecture the global information is available at a centralized Supervisor Module (SM) (we have to mention here that there are two modules, SMa and SMs (SMa is an active supervisor module and SMs is standby supervisor module to be used in case of the failure of SMa)) which takes decisions, resulting in a much flexible and fair access is achieved among users (Interface Cards). In DQDB protocol, media access protocol of the IEEE802.6 standard for MANs, does not fully take advantage of the capacities of a dual bus architecture. In EHABKIT architecture does fully take advantage of the capabilities of a dual bus.

The Fiber Distributed Data Interface (FDDI) is a LAN, the original FDDI-I standard was more suitable for data applications. The FDDI-II standard is suitable for both voice and data applications, the proposed architecture is favorably compares with this too. The FDDI-II operates at data rate of 100Mbps using an optical fiber medium and support about 500 stations, the proposed architecture designed at data rate of 10Mbps using coaxial cable medium and support about 450 stations (ICs) [36,37]. In FDDI a high Target Token Rotation Time (TTRT), causes a long channel access time for stations at high load. This results in a deterioration of real-time traffic behavior of the protocol. The Broadband Integrated Service Digital Network (B-ISDN) relay heavily on an optical fiber technology to bring a wide range of voice, data, image, and video services to customers. The use of multiple optical-fiber taps on a medium introduces significant losses and limits the range of the medium and the number of devices that can be supported [40]. EHABKIT architecture is a single integrated network to be used as a LAN, MAN, and WAN, furthermore, it allows copper and fiber to be intermixed [34]. B-ISDN uses fundamentally the same user interface architecture and the same protocol architecture as ISDN [27,52,53]. The major difference is that B-ISDN supports very high data rate at the user interface. To support these data rate, a new data transfer mechanism is defined known as Asynchronous Transfer Mode (ATM) makes use of fixed size packets, called cells. By using packets of fixed size and fixed format, even greater efficiencies can be achieved with ATM compared to frame-relay. EHABKIT architecture is also, use of fixed size packets. Traffic control in ATM networks is an outstanding problem, which is currently motivating a large research activity. In EHABKIT traffic control can achieved with an appropriate choice of restrictions, it is possible to obtain a balanced response to all traffic types. In an ATM network, all information is packetized and transferred in small, fixed size blocks called cells by using the virtual connection concept. In EHABKIT all information is packetized and transferred in small or large fixed size blocks called packets by using the virtual circuit concept.

In the DATAKIT packet-switch [37], there is a contention bus as opposed to a Scheduler / Data (S/D) bus (used for scheduler slot and to a locate packets on the remaining slots) on which contention is resolved on a slot-by-slot basis. Contention in a slot determines the right for access in the next slot, and all ICs participate in the contention process. As opposed to this in the proposed architecture, information collected in a scheduler slot is used to schedule the next data cycle of n slots. Although, DATAKIT can give a better treatment to higher priority packets and for packet arriving into an idle system, by way of lesser delay, it cannot give any delay bounded for low priority traffic because high priority packets keep winning on contention. This problem is not severe in the proposed EHABKIT architecture because of batch scheduling. With scheduling strategies like the Restricted bound Round Robin [32-36], more balanced scheduling of packets from various traffic types can be expected see Table: III. As in the case of token ring, the DATAKIT cannot be made sensitive to card congestion and other such situations; a flexibility available in the proposed architecture. For instance, a longest queue first kind of scheduling policy within the priority class would sensitive to card congestion. DATAKIT depends upon switch and Common Control Processor (CCP) for scheduling, as the failure of the switch or CCP will cripple the DATAKIT as a whole. This problem is solved in EHABKIT architecture by using two Supervisor Modules one in an active mode (SMa) and the other in hot standby (SMs), as the failure of SMa, SMs can be used as s backup. Both DATAKIT and EHABKIT result in perfect scheduling protocol on the broadcast bus in DATAKIT, and on S/D bus and Data bus in EHABKIT at high Loads, and thus, stable with respect to increasing load. In the way, EHABKIT is as the DATAKIT switch is not synchronous like a time division switch, so an idle Interface Card (IC) consume no bandwidth. In that respect, internally EHABKIT as the DATAKIT is more like a packet switch than a circuit switch even though externally it appears to do circuit switching.

In general, the concept of pure packet switching for EHABKIT architecture is almost equivalent to the DQDB, FDDI, ATM for B-ISDN.

VI. CONCLUSION

The paper discuss comparative performance of high-speed networks carrying multimedia. The proposed EHABKIT high-speed network can support a wide range of applications through short high-speed dual bus and flexible multimedia communication capabilities. Multimedia includes, real-time synchronous voice traffic and multi-priority asynchronous data traffic types. Bounded restriction of higher priority traffic types improves the response of lower priority traffic types at a moderate expense of response to the former. With an appropriate choice of restrictions, it is possible to obtain a balanced response to the real-time synchronous voice and asynchronous data traffic types (Interactive Data and Non-Interactive Data (ID and NID respectively)). Under the Restricted Voice & Data Round Robin (RVDRR) scheduling service strategy it is ensure fair sharing of the transmission capacity, obtain high throughput and network utilization together with low and bound delays, support priorities and different traffic classes, strive for simplicity, robustness and easy implementability, all at the time. In fact these are the challenge of the proposed MAC protocol with a centralized perfect scheduling. The results indicate that the proposed EHABKIT architecture is indeed stable up to fairly high loads and under integrated multi-priority traffics load. It does provide quick delivery of voice packet, while handling fairly large data traffic load as well. Though in general a centralized control architecture is considered inefficient, the proposed EHABKIT with concurrent short high-speed dual bus and its centralized perfect scheduling policy is promising enough. In general, the concept of pure packet switching for EHABKIT architecture is almost equivalent to the DQDB, FDDI and ATM for B-ISDN.

REFERENCES

[1] M.Boissean, et. al., “High Speed Networks,” John Wesley & Sons, 1995.

[2] Thomas E. Anderson, et. al., “High Speed Switch Scheduling For Local Area Networks,” Digital Equipment Corporation, No.99, Palo Alto, California, April 26, 1993.

[3] M. Schroeder, et. al., “Autonet: A High Speed Self-Configuration Local Area Network Using Point-to-Point Links,” IEEE J. SAC, Vol.9, No.8, Oct. 1991.

[4] R. Metcalfe, and D. Boggs, “ Ethernet: Distributed Packet Switching For Local Computer Networks,” Communication of the ACM, Vol.19, No.7, July, 1976.

[5] American National Standards Institute, Inc. Fiber Distributed Data Interface (FDDI), Token Ring Media Access Control (MAC). ANSI Standard X3.139, 1987.

[6] American National Standards Institute, Inc. Fiber Distributed Data Interface (FDDI), Token Ring Physical Layer Protocol (PHY) . ANSI Standard X3.148, 1988.

[7] M. Schroeder, et. al., “ Performance of Firefly RPC,” ACM Transactions on Computer Systems, Vol.8, No.1, Feb. 1990.

[8] B. W. Abeysundara, and A. E. Kamal, “ High Speed Local Area Networks and Their Performance: A Survey,” ACM Computing Surveys, Vol.23, No.2, June 1994.

[9] Zakaria Bin Ali, Mustaffa Samad, Habibah Hashim, " Performance comparison of video multicasting over Asynchronous Transfer Mode (ATM) Multiprotocol Label Switching (MPLS) networks," 2011 IEEE International Conference on System Engineering and Technology (2011)

[10] Han Zhou Han Zhou, C H Chang, " A dynamic bandwidth allocation scheme for multimedia data over ATM networks," Proceedings of International Conference on Network Protocols (1995).

[11] M Ashibani, D Mashao, B Nieya, R Sewusenkar, " Performance evaluation of dynamic bandwidth allocation scheme for VBR video streaming in ATM networks," IEEE AFRICON 6th Africon Conference in Africa (2002).

[12] H A H Kasdirin, R Ab Rahman, "The process flow and analysis of voice over ATM in common communication network,"_4th National Conference of Telecommunication Technology 2003 NCTT 2003 Proceedings (2003).

[13] W M Moh, Yu-Jen Chien Yu-Jen Chien, Teng-Sheng Moh Teng-Sheng Moh, J Wang, Yi-Wen Yang Yi-Wen Yang, " Multiple access protocols for multimedia ATM traffic," Volume: 2, Proceedings of 1994 37th Midwest Symposium on Circuits and Systems (1994).

[14] J M Ng, T Y Liang, E Chan, C F Au, "Performance model of bandwidth allocation strategies for multimedia FDDI networks ,"_Volume: 2, Proceedings of ICCS 94 (1994).

[15] Cheng-Shong Wu Cheng-Shong Wu, Cheng-Chun Yang Cheng-Chun Yang, Jyh-Horng Wen Jyh-Horng Wen, " Bandwidth allocation for FDDI token ring with interworking unit," Volume: 2, Proceedings of ICCS 94 (1994).

[16] S Kamat, N Malcolm, W Zhao, " Performance evaluation of a bandwidth allocation scheme for guaranteeing synchronous messages with arbitrary deadlines in an FDDI network,"_1993 Proceedings RealTime Systems Symposium (1993).

[17] T B Maples, T Suda, " Voice and data integrated transmission on an FDDI network," Conference Record GLOBECOM 92 Communications for Global Users IEEE (1992).

[18] F Tari, V S Frost, "Performance comparison of DQDB and FDDI for integrated networks," 1991 Proceedings 16th Conference on Local Computer Networks (1991).

[19] S Ghani, M Schwartz, "Comparison of DQDB and FDDI MAC access protocols," 1991 Proceedings 16th Conference on Local Computer Networks (1991).

[20] A Fraser, " Towards a Universal Data Transport System," IEEE Journal on Selected Areas in Communications (1983), Volume: 1, Issue: 5, 1983.

[21] Harmen R. Van As, "Media Access Techniques: The Evolution Towards Terabit /s LANs and MANs," Computer Networks and ISDN Systems, Vol.26, No.6-8, March, 1994.

[22] "Distributed Queue Dual Bus (DQDB)," Subnetwork of a Metropolitan Area Network (MAN), IEEE Std. 802.6 / D15, Dec. 1990

[23] M. N. O. Sadiku, and A. S. Arvind, "Annotated Bibliography on Distributed Queue Dual Bus (DQDB) ," Computer Communication Review, ACM SIGCOMM, Vol.24, No.1, January, 1994.

[24] P. Davids, et. al., " FDDI: Status and Perspectives," Computer Networks and ISDN Systems, Vol.26, No.6-8, March, 1994.

[25] W. Stallings, " Handbook of Computer Communication Standards," Vol.2, Howard W. Sama & Company, 1989.

[26] "FDDI Token Ring Media Access Control (MAC)," ANSI X3T9.5, Doc., X3.139 /1987 (ISO 9312/1989).

[27] W. Stallings, " Networking Standards A Guide to OSI, ISDN, LAN, and MAN Standards," Addison Wesley Publishing Company, 1993.

[28] Ehab A. Khalil, and Ayman El-Sayed, "Performance Evaluation of Video/Voice/Data Integration Over VP-Based ATM Ring Network," Accepted For Publication in the 6th Asia–Pacific Conference on Communications (APCC2000), Seoul, Korea, 30Oct. – 2Nov. 2000.

[29] Ehab A. Khalil, and Ayman El-Sayed, "Performance Evaluation of Video/Voice/Data Integration Over VP-Based ATM Ring Network," Published in the Telecommunication and Information Management Journal (TIMJ), P.O. Box 99215 – Pittsburgh, PA 15233, USA, in Vol.2, Issue 4, January 2002.

[30] M. Prycker, "Asynchronous Transfer Mode: Solution For Broadband ISDN," New York Ellis Harwood, 1991.

[31] J. Bellamy, "Digital Telephone," 2nd Ed. N.Y. Wesley, 1991.

[32] Ehab A. Khalil, "Networking Using EHABKIT High Speed Network," Accepted for Publication in the Telecommunication and Information Management Journal (TIMJ), P.O.Box 99215 – Pittsburgh, PA 15233, USA, in April 2001, and appeared in the Vol.3, Issue2, April 2002.

[33] Ehab A. Khalil, "EHABKIT: Towards a Universal Multimedia Transport System," Accepted for Publication in the Telecommunication and Information Management Journal (TIMJ), P.O. Box 99215 – Pittsburgh, PA 15233, USA, in April 2001, and appeared in the Vol.3, Issue2, April 2002.

[34] Ehab A. Khalil, "Fair MAC for EHABKIT Dual Bus Multiple Access High Speed Network (HSN)," Proceeding of the IASTED Intel. Conf. Parallel and Distributed Computing and Networks (PDCN'97), Singapore, August 11-13, 1997.

[35] Ehab A. Khalil, "EHABKIT LAN / MAN / WAN Architecture and Protocol," Accepted for publication in the IASTED Intel. Conf. of Networks, Orlando, Florida, USA, January 8-10, 1996 and published in the Proceeding of the IASTED Intel. Artificial Intelligence, Expert Systems and Neural Networks, Honolulu, Hawaii, USA August 19-21, 1996.

[36] Ehab A. Khalil, "Performance Evolution of EHABKIT LAN/WAN Toward Multimedia Environment," Proceeding of IASTED/ISMM Intel. Conf. Distributed Multimedia Systems and Applications, Stanford University, Stanford California, USA, August 7-9, 1995.

[37] A. G. Fraser, " Towards a Universal Data Transport System," Advances in Local Area Networks, IEEE Press, N.Y., 1987.

[38] Ehab A. Khalil, "A Centralized Bus Based LAN Architecture For Integrated Voice / Data," Ph.D. Thesis, Dept. of Computer Science & Engineering, Indian Institute of Technology (IIT), Bombay-76, INDIA, 1993.

[39] Ehab A. Khalil, and et. al., "Data Traffic in a New Centralized Switch Node Architecture for LAN," Computer Communication Review (ACM) SIGCOMM, USA, Vol.24, No.1, January, 1994.

[40] Ehab A. Khalil and Gamal M. Ali, " A Proposal Modifications to the FDDI's MAC for Increasing Its Performance," Accepted for Publication in the 6th Asia – Pacific Conference on Communications (APCC2000), Seoul, Korea, 30Oct. – 2Nov. 2000.

[41] Ehab A. Khalil and Gamal M. Ali, " A Proposal Modifications to the FDDI's MAC for Increasing Its Performance," Accept for publication in Telecommunications and Information Management Journal (TIMJ), P.O Box 99215 – Pittsburgh, PA 15233, USA, in April 2001, and appeared in the Vol.3, Issue, April 2002.

[42] Ehab A. Khalil, "High Throughput of Asynchronous Traffic and Maximum Capacity of Synchronous Sources with the New FDDI's MAC Protocol, " Accepted in the IJESET, Vol. 1, Issue 2, pp. 27-36, Feb. 2012.

[43] Ehab A. Khalil, "Performance Evolution of High Speed Network Carrying Multimedia," Accepted in the IJAET, Vol. 3, Issue 1, pp.371-380, March 2012.
[44] Ehab A. Khalil, "Simulation and Analysis Studies To Explore The Parameters That Affect The Performance of The FDDI at Gigabit Speed," Accepted in the IJAET, Vol. 3, Issue 2, pp.520-570, May 2012.
[45] Ehab A. Khalil and Gamal M. Ali, "Evolution of a High Speed Network Carrying Video and Data Traffic With an Excellent Proposal," Accepted for Publication in the 6th Asia – Pacific Conference on Communications (APCC2000), Seoul, Korea, 30Oct. – 2Nov. 2000.
[46] Ehab A. Khalil and Gamal M. Ali, "Evolution of a High Speed Network Carrying Video and Data Traffic With an Excellent Proposal," Accept for publication in Telecommunications and Information Management Journal (TIMJ), P.O. Box 99215–Pittsburgh, PA 15233, USA, in July 2001, and appeared in the Vol.3, Issue, July 2002.
[47] Ehab A. Khalil, and Ayman El-Sayed, " Control Mechanism For Fairness Among Traffic On ATM Network ," Published in the Telecommunication and Information Management Journal (TIMJ), P.O. Box 99215 – Pittsburgh, PA 15233, USA, in Vol.2, Issue 1, Spring 2000
[48] Ehab A. Khalil, and Ayman El-Sayed, " Control Mechanism For Fairness Among Traffic On ATM Network ," Accepted For Publication in the 18th IASTED International Conference AI'2000, Austria, Feb.14-17, 2000.
[49] Ehab A. Khalil, "Evaluation of Integrated Multimedia Traffic Over ATM Network," Accepted in the IJESET, Vol. 1, Issue 2, pp.108-117, Feb. 2012.
[50] Ehab A. Khalil, "Integration of Voice and Data in ATM Ring Network," Accepted in the IJAT, Vol. 3, Issue, 2, pp.129-139, May 2012.
[51] B. Mukherjee and C. Bisdikian, "A Journey Through The DQDB Network Literature," Performance Evaluation, Vol.16, No.1-3, 1992.
[52] S. L. Sutherland and J. Burgin, "B-ISDN Internetworking," IEEE Communication Magazine, Vol.31, No.8, August, 1993.
[53] S. Sakata "B-ISDN Multimedia Workstation Architecture," IEEE Communication Magazine, Vol.31, No.8, August, 1993.

Ehab Aziz Khalil, (B.Sc'78 – M.Sc.'83 – Ph.D.'94), Ph.D. in Computer Network and Multimedia in the Dept. of Computer Science & Engineering, Indian Institute of Technology (IIT) Bombay-400076, India in July 1994, Research Scholar from 1988-1994 with the Dept. of Computer Science & Engineering, Indian Institute of Technology (IIT) Bombay-400076, India, M.Sc in the Systems and Automatic Control, Faculty of Electronic Engineering, Minufiya University, Menouf – 32952, EGYPT, Oct. 1983, B.Sc. in the Dept. of Industrial Electronics, Faculty of Electronic Engineering, Minufiya University, Menouf – 32952, EGYPT, May 1978. Since July 1994 up to now, working as a Lecturer, with the Dept. of Computer Science & Engineering, Faculty of Electronic Engineering, Minufiya University, Menouf – 32952, EGYPT.. Participated with the TPC of the IASTED Conference, Jordan in March 1998, and With the TPC of IEEE IC3N, USA, from 2000-2002. Consulting Editor with the "Who's Who?" in 2003-2004. Member with the IEC since 1999. Member with the Internet2 group. Manager of the Information and Link Network of Minufiya University, Manager of the Information and Communication Technology Project (ICTP) which is currently implementing in Arab Republic of EGYPT, Ministry of Higher Education and the World Bank. Published more than 95 research papers and article reviews in the international conferences, Journals and local newsletter.
For more details you can visit http://ehab.a.khalil.50megs.com or http://www.menofia.edu.eg